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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/042,880	01/09/2002	Nicola R. Chong-White	03493.00293	5289

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EXAMINER	
PIERRE, MYRIAM	
ART UNIT	**PAPER NUMBER**
2654	

DATE MAILED: 02/28/2005

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)	
	10/042,880	CHONG-WHITE ET AL.	
	Examiner	Art Unit	
	Myriam Pierre	2654	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-24* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-24* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date *05/09/2002*.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

1. **Claims 1, 10-11, & 16-21 are rejected under 35 U.S.C. 103(a) as being obvious over Coorman et al. (2002/0143526) in view of Sanneck et al. (IEEE GLOBECOM-96).**

As to claim 1

Coorman et al. teach

(a) performing syllable segmentation (syllable segment) on a frame of the speech signal **(Time Scale Modification via slicing speech into short frames of speech, page 8, paragraph 102)** and suggests detecting syllable **(suggestion cited on page 1, paragraph 8)**;

(d) blending **(rejoins)** overlapping segments **(WSOLA, waveform similarity overlap-add, concatenates two segments using overlap-add, & TSM rejoins the segments after repositioning, page 1 paragraphs 2 & 7; & page 4 paragraph 48)**.

Coorman et al. teach TSM, but do not explicitly teach a time scaling factor.

However, Sanneck et al. teach

(b) & (c) applying a scaling factor, which necessarily was determined to the segment in order to modify a time scaling **(time scale modification)** to the segment **(the time scale expansion applies a scaling factor to a segment, thus modifying the time scale used in a segment, page 49, left col., 5th paragraph).**

At the time of the invention, it would have been obvious to one of ordinary skill in the art to apply a determined scaling factor to a segment in order to modify the input signal by time-scale expanding or compressing parts of the signal for improving signal quality.

Coorman et al. also do not explicitly teach retaining frequency attributes.

However, Sanneck et al. teach

retain a frequency attribute **(pitch frequency preserved, page 49, left col., 4th paragraph)** of the speech signal that is processed **(speech signal, page 50, right col., 3rd paragraph)**.

At the time of the invention, it would have been obvious to one of ordinary skill in the art to use TSM, while retaining a frequency attribute in a speech signal in order to preserve natural sound, as taught by Sanneck et al. **(page 49, right col., 3rd paragraph)**.

As to claim 10

Coorman et al. suggests

step (d) utilizes an algorithmic technique **(page 1, 2nd paragraph)** selected from the group consisting of an overlap-add (OLA) technique and a waveform similarity

overlap-add (WSOLA) technique **(speech segment part of modification algorithm, WSOLA and OLA, suggestion cited on page 1, paragraph 7 and page 3 paragraph 32)**.

As to claim 11

Coorman et al. teach

adding overlapping segments **(overlap add procedure for segment concatenation, page 3, 3rd paragraph2)**.

Coorman does not explicitly teach correlating and thresholding.

At the time of the invention, it would have been obvious to one of ordinary skill in the art to correlate overlap segments to a particular threshold in order to effectively ensure that the segments that are added are most similar to each other, thus maintaining the spectral properties of the original signal.

Neither Coorman et al. nor Sanneck et al. explicitly teach retaining the segment if correlation between the two segments is less then the threshold.

However, Official Notice is taken that it would have been obvious to one of ordinary skill in the art at the time of invention that one necessarily would use a correlating threshold to determine the best matched segments to retrain in order to avoid an unnecessary overlap-add process that would result in repeating a sound energy burst.

As to claim 16

Coorman et al. does not explicitly teach thresholding time delay.

However, Sanneck et al. teach

(e) determining a time delay associated with the segment **(time-domain, minimum extra delay, page 48, right column, 1st & 2nd paragraph).**

At the time of the invention, it would have been obvious to one of ordinary skill in the art to determine a time delay associated with a segment in order to cope with packet delay jitter, as taught by Sanneck et al., page 50, right column, 1st paragraph.

Coorman et al. does not teach adjusting the scaling factor of a segment if the time delay is greater then a threshold response.

At the time of the invention, it would have been obvious to one of ordinary skill in the art to adjust the scaling factor of a segment if a time delay is greater then the determined threshold response in order for future compression, thus maintaining the perceived effect of real-time characteristics of the processed speech signal.

As to claim 17

Coorman et al. teach

frequency attribute **(waveforms)** is a short-term **(short-time)** Fourier Transform of the speech signal **(STFT, short-time Fourier Transforms, page 5, paragraph 51, waveforms, Abstract).**

As to claim 18

Coorman et al. suggest

(e) outputting a processed speech signal to a telecommunications network in response to step (d) (by citing in "*High-quality speech output generation through advanced phrase concatenation*" of Workshop on Speech Technology in the Public Telephone Network: Where are we today?, col. 1, paragraph 8).

At the time of the invention, it would have been obvious to one of ordinary skill in the art to output speech signals to a telecommunications network for improved speech quality, thus allowing users to communicate via modern telephonic devices.

As to claim 19

Coorman et al. teach

(e) estimating a pitch component **(pitch period)** of the speech signal **(take the maximum local pitch period of the first segment and the local pitch period of the second segment, page 8, paragraph 108. Implies an estimation of the pitch period)**;

(g) outputting a processed signal to a speech coder **(CEPC)** in response to step (f) **(suggested by max peak of residual speech signal obtained after LPC inverse filtering, page 9, paragraph 113)**.

Coorman does not explicitly teach utilizing the estimated pitch component in order to retain frequency attributes.

At the time of the invention, it would have been obvious to one of ordinary skill in the art to utilize the estimated pitch components used to retain frequency attributes in order to emulate the natural sound of the original speech signal.

As to claim 20

Coorman et al. do not teach using the rest of the recited coders.

However, Sanneck et al. teach PCM (page 50, right col., 1st paragraph).

Official Notice is taken that it would have been obvious to one of ordinary skill in the art at the time of invention that one necessarily would use the standard coders as cited in claim 20 to digitize speech on a sample-by-sample basis because of the coder's ability to output signals which closely match input speech signals.

At the time of then invention, it would have been obvious to one of ordinary skill in the art to choose any of the coders, **in order** to obtain a high level of perceptual quality speech from the standard list of coders for fulfilling the possible applicant/user needs.

As to claim 21

Neither Coorman et al. nor Sanneck et al. teach outputting speech signal to a speech coder.

However, Official Notice is taken that it would have been obvious to one of ordinary skill in the art at the time of invention that for compressing digitized speech

signal for speech transmission or storage, one needs a speech coder for outputting a speech signal.

2. **Claim 2 is rejected under 35 U.S.C. 103(a) as being obvious over Coorman et al. (2002/0143526) in view of Sanneck et al. (IEEE GLOBE-96), as applied to claim 1, and in further view of Coorman (6,665,641).**

As to claim 2

Coorman et al.'s 26 teach
time-scale modification **(TSM)** of a syllable **(page 1, paragraph 8).**

Neither Sanneck nor Coorman et al. specifically teach consonant – vowel transition and a steady-state vowel syllable.

However, Coorman et al. (6,665,641) teach concatenation of consonant – vowel transition (consonant-vowel transitions, page 3, line 14) and a steady-state vowel (conservation of vowel sound, page 3, line 14).

At the time of the invention, it would have been obvious to one of ordinary skill in the art to use a unit of speech sound, such as syllabic consonant steady state vowels and vowel transitions for a mixed inventory of speech units for synthesis in order to increase speech quality, as taught by Coorman et al (6,665,641), **(col. 2, line 32)**.

3. **Claims 3-4 are rejected under 35 U.S.C. 103(a) as being obvious over Coorman et al. (2002/0143526) in view of Sanneck et al. (IEEE GLOBE-96), as applied to claim 1, in further view of Coorman (6,665,641), as applied to claim 2, and in further view of** Melanson et al. (6104822).

As to claims 3 and 4

Coorman et al. do not teach time *expansion*.

However, Sanneck et al. time *expansion* (**time scale expansion applies a scaling factor to a segment, thus modifying the time scale used in a segment, page 49, left col., 5th paragraph**).

At the time of the invention, it would have been obvious to one of ordinary skill in the art to use time expansion for TSM for syllables in order to emphasis the parts of a syllable that is harder to hear, such as the consonants, thus providing speech intelligibility.

Neither Coorman et al. nor Sanneck et al. nor Coorman et al. (6,665,641) teach time *compression*.

However, Melanson et al. teach syllable rate compressor (syllabic rate compressor, col. 2, lines 57-60).

At the time of the invention, it would have been obvious to one of ordinary skill in the art to use time compressing for syllables in order to limit the gain of a loud vowel sound while amplifying a soft consonant that immediately follows it for applications in hearing aides, as taught by Melanson, col. 2, lines 57-60.

4. Claims 5-6 are rejected under 35 U.S.C. 103(a) as being obvious over Coorman et al. (2002/0143526) in view of Sanneck et al. (IEEE GLOBE-96) and Coorman (6,665,641), as applied to claim 2, in further view of Covell et al. (5,828,994).

As to claim 5

Neither Coorman et al. nor Sanneck et al. teach time compression occurs during low energy regions of the speech signal.

However, Covell et al. teach time compression occurs during low energy regions of the speech signal (**compress pauses, which are the low amplitude portion of the speech signal, col. 5, lines 19-20 & 22-27**).

At the time of the invention, it would have been obvious to one of ordinary skill in the art to compress a speech signal during low energy levels, especially pauses, in order to capture the relative differences between stressed, unstressed, and pause sounds, as taught by Covell et al., col. 6, lines 12-19.

As to claim 6

Coorman et al.'s 26 teach

wherein a time duration of the speech signal is essentially equal to a time duration of the processed speech signal **(real-time time domain concatenation of digital speech waveforms, and TSM, page 1, paragraphs 2 & 6)**.

5. Claims 12-15 are rejected under 35 U.S.C. 103(a) as being obvious over Coorman et al. (2002/0143526) in view of Sanneck et al. (IEEE GLOBECOM-96).

As to claim 12

Claim 12 is rejected for the same or similar reasons as claim 5 above.

At the time of the invention, it would have been obvious to one of ordinary skill in the art to detect a high energy region of the speech signal because of the need to emphasize the difference between an energy signal that represents consonants and vowels; thus, if one is detecting low energy levels, as in claim 5 above, the higher energy regions are detected as well.

As to claim 13

Coorman et al. does not teach

detecting changes in frequency domain characteristics.

However, Sanneck et al. teach

detecting abrupt changes in frequency-domain characteristics of the speech signal **(test environment of 4 speech signals with different pitch frequencies, page 50 left column, 3rd paragraph)**.

At the time of the invention, it would have been obvious to one of ordinary skill in the art to detect abrupt changes in frequency domain characteristics in order to judge the presence of disturbance components, such as "tinny, metal" or "interrupted, clicking", as taught by Sanneck, **page 50, right column, 3rd paragraph.**

As to claim 14

Coorman et al. teach

cross-correlation measures **(cross-correlation, page 4 paragraph 43).**

As to claim 15

Coorman et al. does not explicitly teach

Amplifying (maximize) a first portion of the TSMS in order to partially restore an associated energy in response to step (c) **(maximization of the cross-energy term, page 4 paragraph 45)**.

At the time of the invention, it would have been obvious to one of ordinary skill in the art to amplify the first portion of the time-scale modified syllable because of the need to emphasize the difference between an energy signal that represents a consonant, which is harder to hear, and an energy level of an adjacent vowel, which is easily recognizable.

6. **Claims 7-8 are rejected under 35 U.S.C. 103(a) as being obvious over Coorman et al. (2002/0143526) in view of Sanneck et al. (IEEE GLOBE-96), as applied to claim 1, in further view of Kates et al. (2003/0072464).**

As to claim 7

Coorman et al. teach frequency domain processing **(speech segments can be concatenated in frequency domain, page 1, paragraph 3)**

Neither Coorman et al. nor Sanneck et al. explicitly teach modifying the frequency domain characteristics to enhance acoustic clues.

However, Kates et al. teach

modifying frequency domain characteristics of the speech signal in order that a transformed speech signal is characterized by enhanced acoustic cues **(warped spectral enhancement used for hearing aid, page 4, paragraph 45)**.

At the time of the invention, it would have been obvious to one of ordinary skill in the art to modify frequency domain characteristics, such as spectral enhancement, in order to improve listening and possibly the level of speech intelligibility, as taught by Kate et al., page 1, 4th paragraph.

As to claim 8

Neither Coorman et al. nor Sanneck et al. teach increasing spectral peaks.

However, Kate et al. teach

adaptive spectral enhancing the speech signal, wherein a distinctness of spectral peaks of the speech signal is increased **(amplify peaks of the spectrum, page 6, paragraph 63)**.

At the time of the invention, it would have been obvious to one of ordinary skill in the art to increase the spectral peaks of a signal because the peaks tend to be more important than the valleys, such as consonants, which are characterized by the regions of maximum spectral power, as taught by Kate, page 6 paragraph 63.

7. **Claim 9 is rejected under 35 U.S.C. 103(a) as being obvious over Coorman et al. (2002/0143526) in view of Sanneck et al. (IEEE GLOBE-96), as applied to claim 1, in further view of Kates et al. (2003/0072464), as applied to claim 8, and in further view of HOU et al. (5,729,658).**

As to claim 9

Neither Coorman et al. nor Sanneck et al. teach about masking.

However, HOU et al. teach

emphasizing higher frequencies of the speech signal **(lower gains at lower frequencies, col. 10, lines 14-16. Implies emphasizing higher frequencies if using lower gains at lower frequencies)**, wherein an upward spread of masking of the speech signal is reduced **(lower gains at low frequencies reduce the upward spread of masking, col. 10, lines 14-16)**.

At the time of the invention, it would have been obvious to one of ordinary skill in the art to emphasize the higher frequencies to reduce speech intelligibility.

8. **Claims 22-24 are rejected under 35 U.S.C. 103(a) as being obvious over Coorman et al. (2002/0143526) in view of Sanneck et al. (IEEE GLOBE-96), as applied to claim 1, in further view of Coorman (6,665,641), as applied to claim 2, in further view of Salmi et al. (5,903,655), as applied to claim 11 above.**

As to claim 22

Coorman et al. teach

*energy corresponding to the frame***(speech segment)** **(shape may differ between the speech segments, energy condition, page 4, paragraph 44.)** *in order to detect a time-scale modification syllable (TSMS)* ***(TSM, page 1 2nd paragraph).***

Neither Coorman et al. nor Sanneck et al. teach

a spectral feature transition rate (SFTR) **(spectral)** *corresponding to the frame* in order to detect a time-scale modification syllable (TSMS).

At the time of then invention, it would have been obvious to one of ordinary skill in the art to analyze the spectral rate corresponding to the TSM of a syllable **in order** to adjust the frequency characteristics of the signal, thus emphasizing the pitch or other features which are not readily corrected in the time-domain.

The rest of the listed the limitations in claim 22 are the same or similar as on claims 1, 4, 8, 9-10, and 16 above, and so are rejected for the same reasons, as follows:

(Adaptive spectral enhancing the speech signal, wherein a distinctness of spectral peaks of the speech signal are increased) and so are rejected for the same reasons, as in claim 8 above.

(Emphasizing higher frequencies of the speech signal, wherein an upward spread of masking of the speech signal is reduced) and so are rejected for the same reasons, as in claim 9 above.

(Outputting a modified frame in response to processing all constituent segments of the frame) and so are rejected for the same reasons, as in claim 21 above.

The rest of the limitations in claim 22 are same or similar as on claims 1, 4, 10 and 16 above, and so are rejected for the same reasons.

As to claim 24

Coorman et al. teach

overlapping techniques **(page 1, paragraph 7)**

(e) overlapping segment **(overlap add representation, page 4 paragraph 48)** that is best-matched to the segment according to a cross correlation **(max of the normalized cross-correlation function, page 4 paragraph 46)** and waveform similarity criterion **(waveform similarity condition, page 4 paragraph 43)** and to the speech component if the frame has a voiced characteristic **(voiced speech waveform, page 4, paragraph 44)**; **&** (f) combining (blending) the segment with an adjacent segment **(waveform blending, first waveform and leading part of the second waveform segment, page 4, paragraph).**

The rest of the listed the limitations in claim 24 are the same or similar as on claims 11, 19, 22, and 24 above, and so are rejected for the same reasons, as follows:

(Estimating the pitch component of the frame) and so are rejected for the same reasons, as in claim 19 above.

(Overlapping and adding the segment and the overlapping segment if a correlation between the segment and the overlapping segment is greater than a threshold) & (essentially retaining the segment if the correlation between the segment and the overlapping segment is less than the threshold) and so are rejected for the same reasons as claim 11 above.

The rest of the limitations in claim 24 are rejected for the same or similar reasons as claim 22 above.

Claim 23 recites the same or similar limitations as claims 22 and 24 rejected above, and so is rejected for the same reasons.

Conclusion

9. The following art made of record and not relied upon is considered pertinent to applicant's disclosure *Roelands, Marc et al.*, (Waveform similarity based overlap-add (WSOLA) for time-scale modification of speech); *Wayman, J.L. et al.*, (High quality

speech expansion, compression, and noise filtering using the sola method of time scale modification); *Wong, P.H.W. et al.* (On improving the intelligibility of synchronized overlap-and-add (SOLA) at low TSM factor); *Covell, M. et al.* (MACH1: nonuniform time-scale modification of speech); *Erogul, O. et al.* (Time-scale modification of speech signals for language-learning impaired children); *Ross, K.N. et al.* (A dynamical system model for generating fundamental frequency for speech synthesis); *Yong, M. et al.* (Study of voice packet reconstruction methods applied to CELP speech coding); Hura (6,026,361); Aguilar et al. (6,691,082); Jacks et al. (4,692,941) Yamada et al. (4,979,212); Bu et al. (2002/0133332); Miller et al. (4,820,059); Tanaka et al. (5,611,018); Tallal et al. (6,413,098); Gersho et al. (6,233,550); Ginzburg et al. (6,285,979); Kurittu et al. (2004/0120309); Goldenthal et al. (5,625,749); Kamai et al. (5,864,812); Li (6,850,577); Sicher et al. (2001/0015968); Goodwin et al. (2003/0093282); Andringa et al. (6,745,155); Nishiguchi et al. (5,752,222); Choi et al. (6,304,843); Soli et al. (6,563,931); Yeldener et al. (5,774,837); Savic et al. (5,327,521), Satyamurti et al. (5,828,995); & Goldberg (5,553,151).

Roelands, Marc et al., teach synchronization criterion for time-scaled modification using overlap –add, also uses WSOLA for speech processing, as well as modified STFT. Wayman, J.L. et al., teach apply time domain modification to difficult speech segments using SOLA, allowing low bit-rate transmission and storage, and improving the overall signal-to-noise ratio.

Wong, P.H.W. et al., teach an algorithm that modifies the SOLA by using a time scaling factor to the input signal for intelligibility.

Covell, M. et al. teach non-uniform time compression, compressing consonants based on stress levels of the neighboring vowels, pauses, silences, and vowels, estimates local energy and controlling segment duration variation to mimic natural speech.

Erogul, O. et al. teach modifying speech signal using different time-scale modification and WSOLA algorithms using time, pitch extractions, and sinusoidal analysis based on frequency and phase of the waves, the system is implemented for language-learning impaired children.

Ross, K.N. et al. teach prosodic patterns from frequency and energy contours that enable parameter labeling of speech signals.

Yong, M. et al. teach speech coding via CELP speech coders implemented in communications network.

Hura teach speech intelligibility using contrasts in consonant and vowel words.

Aguilar et al. teach speech signals using pitch and voice algorithm to parameterize speech signals, codes sub-band signals with waveform coding.

Jacks et al. teach time domain technique for phoneme and transitions, segmenting speech and varying pitch for naturalness of sound, real-time application.

Yamada et al. teach time normalization pattern, segmenting signals that may have unequal length.

Bu et al. teach phonetic speech recognition sampling signals for Fourier transform calculation to determine amplitudes of component waves for stationary and mel-scale

frequency for non-stationary vowels.

Miller et al. teach speech processing via parameter analysis for various speech categories.

Tanaka et al. teach expanding phonemes and changing acoustic spectra and emphasis added to fast changing segments of the phonemes.

Tallal et al. teach voice speed converting device that judge voice and silence section of sound signal, compresses and expansion process included.

Gersho et al. teach a decoder for reproducing speech, classifying speech, including transition portions of speech.

Ginzburg et al. teach phoneme analysis, detecting energies in various classified signals.

Kurittu et al. teach changing size of jitter buffer at receiving end of communications system, time alignment based on speech data.

Goldenthal et al. teach phonetic recognition by capturing behavior of acoustic attributes used to represent a speech waveform.

Kamai et al. teach synthesizing speech segments such as speech waveforms, pitch waveforms and differential waveforms.

Li teaches signal processing that discriminates signals, exchanging voice signals between a switched circuit network, used for transmission, enabling voice and data exchange.

Sicher et al. teach various voice protocols and coders used for speech trans-coding and data inter-working.

Goodwin et al. teach expressing input signal via transform –domain representative with

or without intermediate time-domain conversion.

Andringa et al. teach a basilar membrane model used to receive input signals, analysis based on frequency estimation, period contour estimation, mask formation and parameterization of reconstructing signal for coding recognition.

Nishiguchi et al. teach speech decoding and decoding, gain adjustments and correction due to spectral shaping.

Choi et al. teach linear prediction filtering determining the magnitude spectrum, phase and reconstruction of excitation signal from the deterministic signal, and uses coders such as WI and MELP.

Soli et al. teach auditory prosthesis to filter out unwanted signals, adapting to environmental sound.

Yeldener et al. teach encoding and decoding speech signals, parameter representation of voiced/unvoiced signals, various coders such VSELP, MBE, IMBE, LC, DPCM, and PMC.

Savic et al. teach real time transformation system, segmenting speech into overlapping segments, and excitation content of the source signal.

Satyamurti et al. teach speech compression and time-scale expansion.

Goldberg teaches electronically controlled device for speech processing with applicability in hearing aids and other types of electro-acoustic communication devices.

10. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Myriam Pierre whose telephone number is 703-605-

1196. The examiner can normally be reached on Monday – Friday from 5:30 a.m. - 2:00p.m.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Talivaldis Smits can be reached on 703-306-3011. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

11. Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

02/17/05

RICHEMOND DORVIL
SUPERVISORY PATENT EXAMINER